“Tiny” Audio DSP Core & Tools

and

Oxford Digital EQ

An introduction and overview

Oxford Digital

1. About Oxford Digital Limited

1.1 Background

- Oxford Digital Limited (ODL) is an independent company that span out of Sony's Oxford Pro-Audio Lab in June 2006. Sony has no retained interest in ODL.
- ODL's core strength is in its engineering team of highly experienced audio design engineers with a diverse skills set that provides end-to-end project coverage
- ODL's core businesses:
  - Technology Licensing:
    - Tiny DSP Core and Toolset – an end-to-end audio DSP solution
    - Oxford Digital EQ – for real-time arbitrary frequency response creation
  - Contract Research and Development
    - Problem solving and development of new Intellectual Property for clients
    - Accelerating Time to Market
    - Provision of specialist skills
    - Integration/customisation of Tiny Core and Oxford Digital EQ for client applications

1.2 The ODL Team

Team skills include:

- Processor architecture development
- Firmware and HDL for FPGAs and ASICs
- Maths & signal processing algorithm development
- DSP Effects and Algorithms
- System Architecture/System Level design
- Hardware design
- All types of software (including real-time embedded)
- Core Research and Development
- Collaboration with our Clients and their Customers (where required)

These broad skills allow us to handle all parts of a project (where required) from concept through to customer support for our clients

1.3 Customer Base

Customers who have been kind enough to release us from our mutual NDAs include:

- **D&M Holdings Inc. (USA) [Denon, Marantz, Macintosh, …]**
- SADIE **(UK)**
- **Sony Ericsson Mobile Company (Japan)**
- **Sony Semiconductor (Japan)**
- YAMAHA **Commercial Audio (Japan)**

2 Tiny Core and Tools

2.1 Concept and Unique Selling Points (USPs)

Provision of a “world class” end-to-end solution for audio DSP that provides:

- A comprehensive design environment and toolset
- Very fast work-flow through the use of hierarchical Graphical Programming Environment high-level front-end, Core Synthesis and back-end “Tuning” tools
- Very low gate count solutions for FPGA and ASIC
- Very efficient execution (e.g. only 5 instructions for a biquad filter with saturating arithmetic)
- Compact code: The high level compiler produces code that on average is 10% smaller than that produced by an expert Assembly Level programmer
- A scalable solution in bit-width, sampling frequency, instructions per sample and number of channels
- Automatic software re-use for different configurations of the Tiny Core as no changes are required at the top design level, the compiler produces the required executable code
- Support for encrypted code to allow effects manufacturers (e.g. SRS Labs Inc.) to provide library items that remain private (i.e. it is not possible to push down through the hierarchy to see the internal workings)
- HDL that is easily portable across different FPGAs and ASIC processes and geometries
- Audio DSP Effects library
- Real-time adjustment of parameters (whilst listening) on either evaluation boards or target hardware
- Real-time injection of test signals and ‘scope probe’ type debugging/audio monitoring
- Provision of Test Bench and Test Vectors for the Core including the ability to simulate the DSP output from the Graphical Programming front-end. Semiconductor manufacturers can validate their results by simulating the Tiny core including the desired effect pre- and post- synthesis as part of the signoff process
- Extremely fast Time to Market for both FPGA and ASIC implementations through the use of the Tiny Toolset (e.g. delivery of DSP code for an application together with HDL for a custom core in 6 days)
- Provision of programmability (as opposed to fixed architecture) by use of the Tiny Core de-serialises development of algorithms and silicon and allows substantial overlap in the process, thus reducing time to market even further

2.2 Typical Applications

The Tiny Core is scalable and can be used in numerous applications including:

CE Equipment Sound Quality Improvement

Addition of DSP in a Class-D Amp or ASIC elsewhere can improve the sound quality of:

- Accessory Speakers
- Headphones
- Flat Panel TVs
- Personal and Handheld Devices:
- Cell Phones
- Flash and HDD MP3 Players
- PDAs

- Digital Still Cameras
- Portable Gaming Consoles
- Portable Navigation Systems

by:

- Removing loudspeaker and cabinet resonances — *for a clear and natural sound*
- Extending Bass Frequency Response — *for richer bass*
- Increasing Loudness with low voltage drivers — *for stronger performance*
- Restoration of MP3 compressed music — *for better sound quality*
- Dynamic Range Control — *to assist listening in noisy environments*
- Stereo Widening on devices such as Cell Phones — *to give a wide image*
- Dialogue Processing — *to improve speech intelligibility*

High End Equipment

FPGA solutions for higher end consumer and professional equipment that provide the ultimate in quality DSP audio processing are extremely practical and cost-effective platforms.

2.3 The Tiny Core

The Tiny Core supports stream based processing of time domain audio samples including single cycle multiply-adds, and multiply-subtracts. With built-in support for sample delay memory of varying size and automatically saturating arithmetic instructions, the Tiny core makes it possible to design a biquad filter in five instructions.

Tiny Core is a reconfigurable RTL model that permits synthesis of FPGA and ASIC gate-level netlists with these parameters:

- 1 - 32 audio I/O channels
- data path bit width from 16 to 48 bits (in steps of 2 bits)
- 128 - 8192 instructions per audio sample period to process all I/O channels
- 1 - 3 data storage RAMs each independently containing between 2^5 and 2^12 data words
- optional external memory interface for large delays (e.g. sparse FIR filters, reverbs)
- support for encrypted code (so that effects manufacturers (e.g. SRS Labs Inc.) can allow their effects to be used securely and remain private)

The core makes no assumptions on master clock frequency and so is independent of audio sample rate. It is optimised for low power and low gate count through a configurable memory access method and a split multiplier design.

A Gate Count Calculator is available (under NDA) that allows examination of various options of key parameters

2.4 Graphical Programming Environment

The Graphical Programming Environment (GPE) is a fully hierarchical design tool that allows rapid design and prototyping of DSP and algorithms with the ability to control parameters and hear/measure results in real-time.

The GPE includes:

- **TinyDraw**
  A hierarchical schematic entry tool
- **TinyNice**
  Optimising compiler (18 passes in the blink of an eye)
- **TinyGCon**
  A GUI Control interface that provides real-time parameter adjustment
- **Evaluation Board**
  Both ASIC and FPGA based evaluation boards are available. These allow DSP algorithms and effects to be developed and used in real-time either connected to the development environment or stand-alone in prototype equipment for evaluation purposes

Figures 1 to 3 show some of these features.

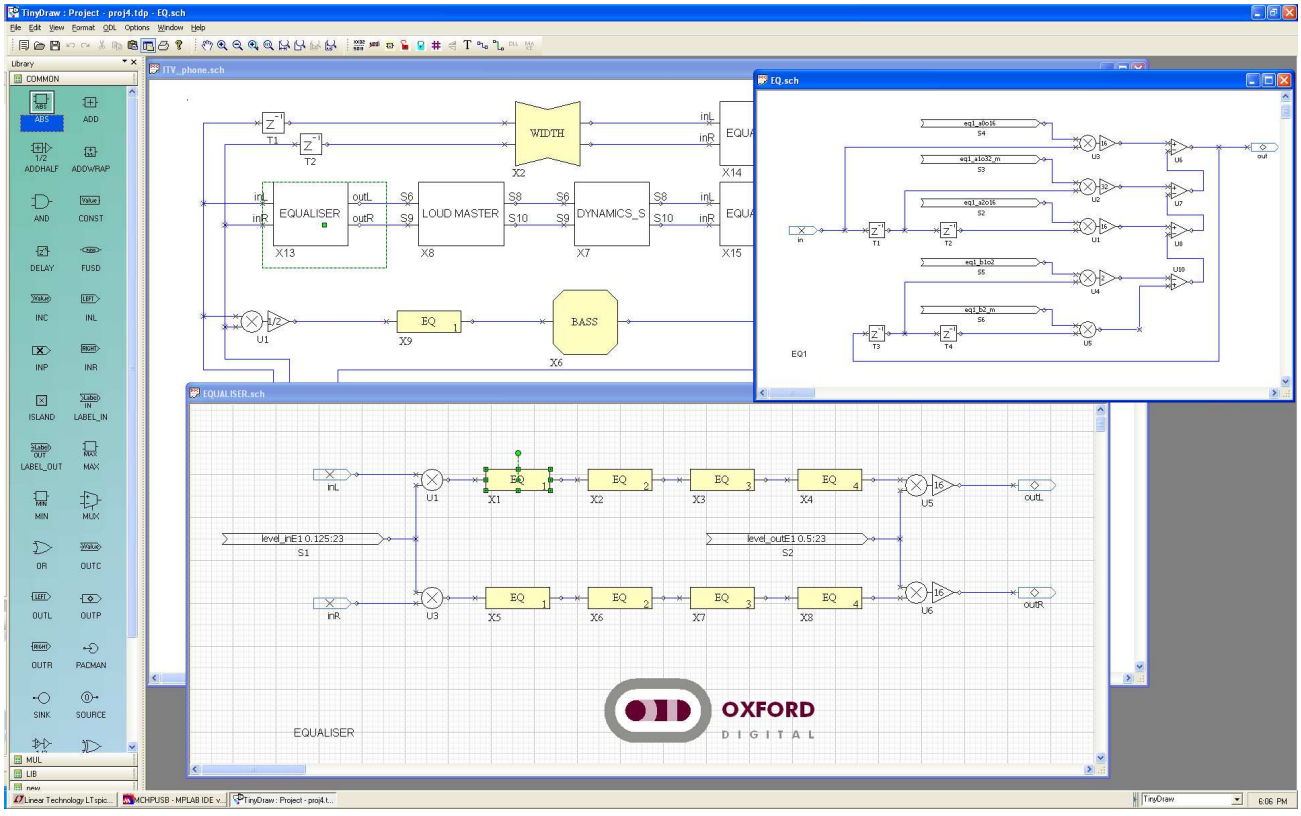

Fig. 1 TinyDraw Screen: Component selection Left, high-level Cell Phone application in background, expansion of EQ block in mid-ground and detail of an EQ element in foreground (3 levels of hierarchy shown here)

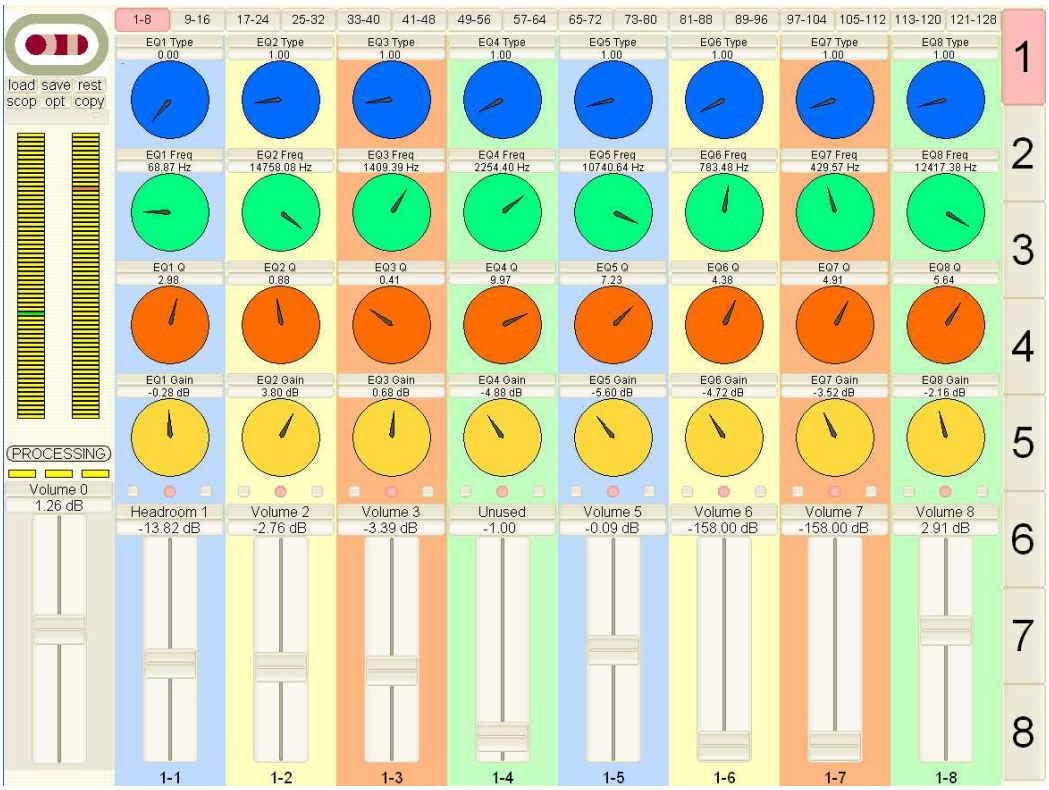

Fig. 2 ***TinyGCon*** GUI Control interface: This shows 1 of the 16 pages of controls that can be assigned and the 8 scene tabs on the RHS which can be used to compare different settings

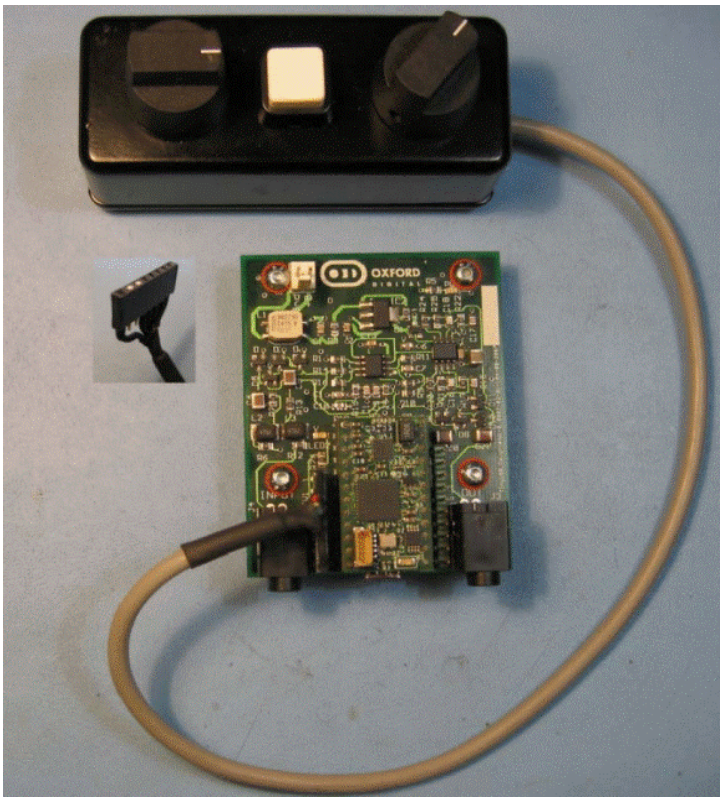

Fig. 3 Evaluation Board with Control Box for Stand-alone operation. This can be embedded in equipment to demonstrate the effect on/off and to choose different ‘scene’ pre-sets.

2.5 Effects Libraries

An Effects Library is available for use with the GPE. It includes effects such as:

- Various Filter and EQ configurations
- Several types of Bass Enhancement that address different speaker deficiencies
- Compressor
- Limiter
- ***LoudMaster*** (an effect to make sounds louder without increasing peak level)
- ***EdgeMaster*** (an effect that modulates transients to add punch (+ve) or reduce transients (-ve))
- Noise Gate

2.6 CE Tuning Tool

This tool allows the sonic performance of CE equipment such as Flat Panel TV, AV, Accessory Speakers, Cell Phones, GPS systems to be improved with the objectives stated in 2.2 above.

The Tuning Tool includes pre-configured DSP that runs on a Tiny Core and includes the following effects:

- Many EQ sections for control of cabinet and speaker resonances
- Two different types of ***BassMaster*** Bass Frequency extension to cope with the different problems inherent in a wide range of equipment
- ***LoudMaster*** which increases subjective loudness without the need to increase peak drive voltage. This is extremely useful in portable and mobile equipment with low PSU voltage
- Effects to restore richness of mp3 and other compressed music
- ***LevelMaster*** which is able to seamlessly adapt the dynamic range to make sounds clear in a noisy environment
- Stereo Widening

The parameters in the DSP are set up for each particular model produced using a simple one page real-time GUI shown in Fig 4.

Once the optimum parameters are identified for a particular model of CE equipment that has a chip incorporating the Tiny Core designed in, the parameters can be exported for inclusion in ROM in the production line.

Training is available to allow engineers with good listening ability to become expert in Tuning the CE equipment with the EasyTune Tuning Tool.

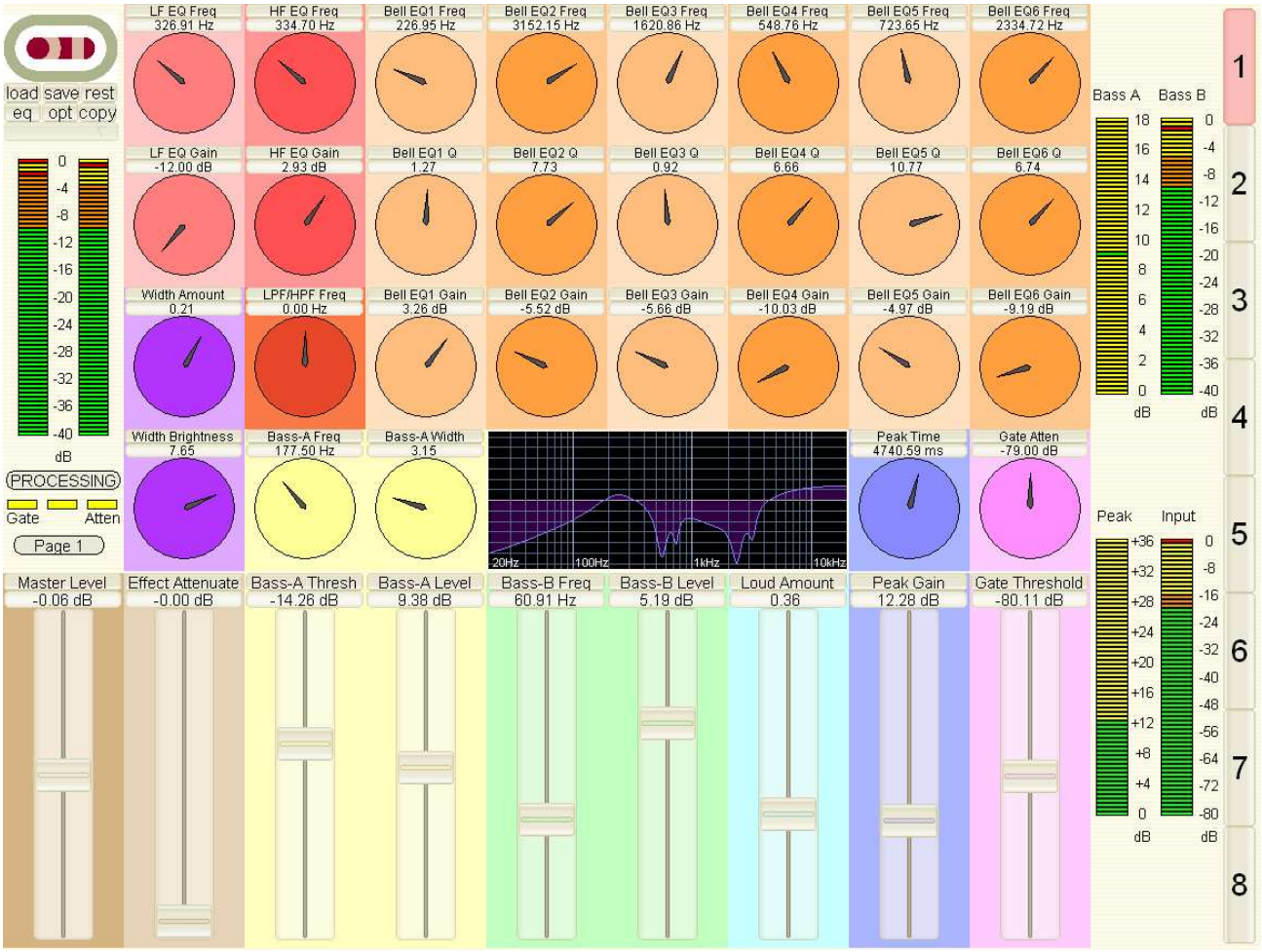

Fig. 4 Tuning Tool GUI – Frequency Display can be enlarged by clicking on it.

3 Oxford Digital EQ

3.1 Concept

The Oxford Digital EQ provides comprehensive EQ facilities including two new, patent protected EQs:

- ***SmoothEQ***
  A new EQ that allows arbitrary specification of frequency response from either GUI or automatic input of frequency and gain at arbitrary points
- New Graphic EQ
  based on a constrained set of SmoothEQ frequency points

3.2 Application Areas and Unique Selling Points (USPs)

Include:

- Automatic correction of CE device resonances (integration with Tiny Tuning Tools)
- Sound / Touring Sound / Installed Sound Venue Correction
- Room Correction (e.g. Home Theatre & AV Systems)

- Music Production (New creative possibilities through GUI and combination of different effects)

There are several techniques already in use for correction of response curves by arbitrarily specified filters which mostly fall into:

- Use of 2nd order Bell EQs
  This is a long, tedious and skilled process as each time a new EQ is added it also has interaction with all other EQs. In addition, most required correction is not symmetrical in shape (unlike Bell EQs) making exact matches difficult to achieve

- Use of FIR Filters
  The response of the equipment, room or venue can be captured by measurement system, then inverted to produce the required corrective response and finally turned into an FIR filter by convolution. Unfortunately this has two weak points:

  o As low frequencies are usually involved, the FIR filter has many taps and the delay through the filter is such that it makes it unusable in many "live" applications where sound latency is an issue

  o There will be a need to adjust the response due to errors and artefacts in the measurement system. It is not possible to make fine adjustments to this type of EQ (which may have 1000s of parameters), so a second layer of EQ and processing needs to be added for correction of these errors

The SmoothEQ does not suffer from any of these problems – see below.

3.3 SmoothEQ

Features

- Minimum phase IIR filters for low delay latency (required in live sound and other areas)
- Dynamically Controllable in Real Time
- No nasty noises when changing response
- Ability to produce better quality results and much faster than conventional EQ methods for arbitrarily specified responses
- Use of less DSP resource than conventional EQ methods for arbitrarily specified responses
- Ability to easily trim results for fine adjustment without adding a new layer of EQ

3.4 Oxford Digital EQ V2.0

Oxford Digital EQ V2.0 includes:

- ***SmoothEQ***
- A New Graphic EQ which operates without interaction of bands
- High and Low-Pass filters with continuously variable slope
- Baxandall Tone Control
- High and Low Shelving filters which have continuously adjustable in-band frequency response 'over' control.
- Classic 'bell' or 'presence' filters with continuously variable gain, frequency and Q

Demonstration versions of Oxford Digital EQ V2.0 are available for download (under NDA).

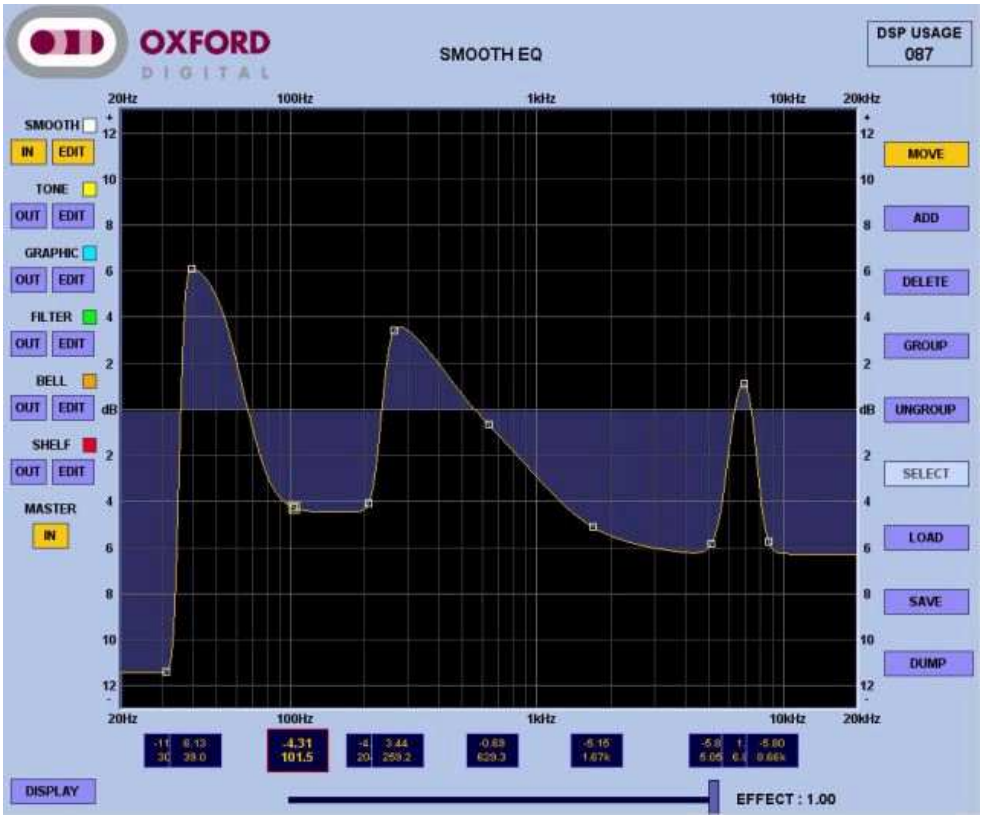

Fig. 5 Example of response of the ***SmoothEQ*** (Yellow line shows actual response)

Fig. 6 Example of response of the New Graphic EQ (Yellow line shows actual response)

4 Marketing Activities

4.1 Technical Papers

Oxford Digital has presented technical papers at several conferences worldwide (see Appendix A).

4.2 Conventions & Exhibitions

Oxford Digital has sponsored and exhibited at many conferences and exhibitions alongside other leading brands – see selection below.

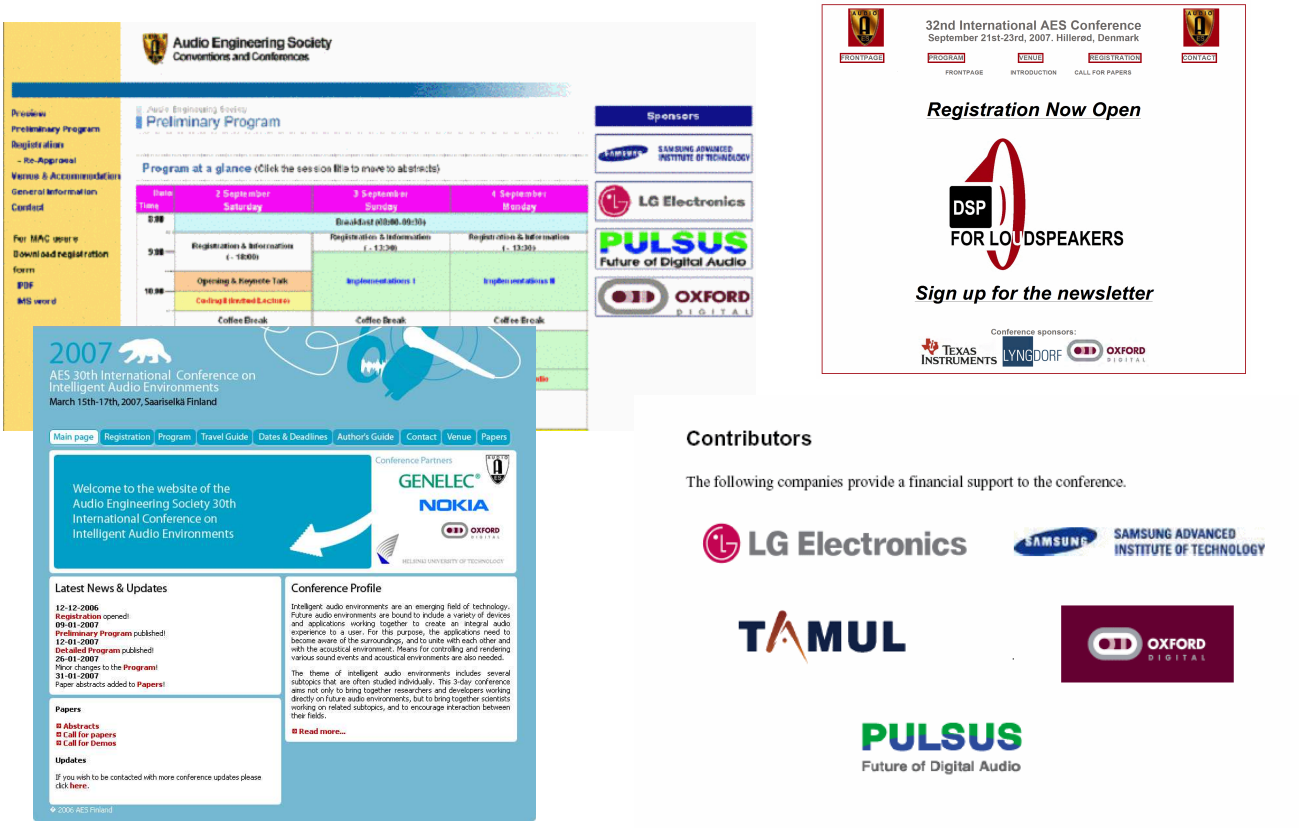

Fig. 7 Some examples of Trade Shows, Exhibitions and Conference Sponsorship

5 Contact Information

We are happy to discuss any of the issues above (or indeed any other audio related issues) in more depth. Please contact:

Name: John Richards, CEO

E: john.richards@oxford-digital.com

T: +44-845-450-5664
F: +44-845-450-5663

W: www.oxford-digital.com

APPENDIX A

Selected Oxford Digital Technical Papers

1. P. Eastty, “Digital Audio Processing on a Tiny Scale: Hardware and Software for Personal Devices”, Paper Number 7207, AES 123rd Convention, New York, October 2007.

2. N. Bentall, P. Eastty and D. Stott, “An Efficient, Low-Noise Filter Architecture for Bass Processing on a DSP Core”. Paper Number 7351, AES 124th Convention, Amsterdam, May 2008.

3. N. Bentall, P. Eastty and D. Stott, “Tiny DSP: DSP Core, Algorithm Development and 'Device Mastering'”. Paper Number 6 AES 34th International Conference: New Trends in Audio for Mobile and Handheld Devices, Jeju Island, South Korea, August 2008.

4. P. Eastty, “Accurate IIR Equalisation to an Arbitrary Frequency Response, with Low Delay and Low Noise Real-Time Adjustment”, Paper 7639, AES 125th Convention, San Francisco, October 2008.